CASE STUDY

CALL RECEPTION IN YEALINK SIP TELEPHONE EXTENSIONS

Description

This document describes the way to setup MEET IP intercom system from FERMAX to communicate with IP YEALINK Telephone by using SIP protocol.

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INTRODUCCIÓN

In this Case Study, it is described the way to integrate MEET IP Intercom with YEALINK SIP telephones.

Despite it is possible to make a phone call to these telephones through a SIP server, in this integration case we will focus in making a call peer to peer directly from the intercom to the telephone, without the need of a SIP server. In case the telephone is registered in a SIP server, it is adviced to review the relative case study of the SIP SERVER in MEET WORKS WITH repository meet.fermax.com

This integration case is focused in the YEALINK TG40 Telephone. With other YEALINK telephones, the process is similar, specially what is related to MEET IP intercom. For more info related to the SIP phone, it is possible to know more at YEALINK SERVICE HUB https://service.YEALINK.com , looking for the FAQ "How To Make Peer-to-Peer IP Calls without a Registrar"

In this case, the configuration to be done will be on a 1 pushbutton MILO MEET intercom panel. Other keypad MEET intercom panels are also compatible with SIP Call.

VERSIONS USED :

- YEALINK TG40-SIP 10.1.57.14
- MILO MEET 1 pushbutton panel v3.0

Direct call is often used in those cases in which due to proximity reasons, the intercom panel and the phone are in the same LAN network and the SIP server is in the cloud. It is not necessary to have internet access to do the call from the intercom to the phone. In addition, the call is faster and reliable since even in the case the internet access is not available, the call will be produced since intercom and telephone are in the same LAN.

As an additional remark, despite we will focus in making a call to a single SIP phone, the intercom outdoor **panel can call simultaneously several destinations or telephones**, following a similar method.

PRELIMINARY CONSIDERATIONS

Before proceeding to the system configuration, it is required to take into account the following points:

IP intercom panel and telephone addressing:

Both, telephone and intercom, need a fixed IP, in the same local IP network where they are installed. Obviously, these IP addresses must not be used in other IP devices. Can be requested to the network admin. For this case, we will use will the YEALINK ip address 192.168.1.100 and the MEET MILO intercom 192.168.1.51 ip address.

OUTDOOR PANEL INTERCOM SETUP

First of all, all devices should be setup in the same ip network range, as per the selected ip addresses.

We will access to the intercom web server. In the MILO MEET 1 pushbutton panel the default ip address is 10.1.1.2 (in MEET Digital panels with keypad like KIN, MILO, MARINE the default address is 10.1.0.1). We must put the computer network interface in the same range 10.1.1.x

The recommended web browser to use is **Google Chrome**. We will introduce the default IP 10.1.1.2 and the user / password to access. By default **user: admin pass: 123456 .**

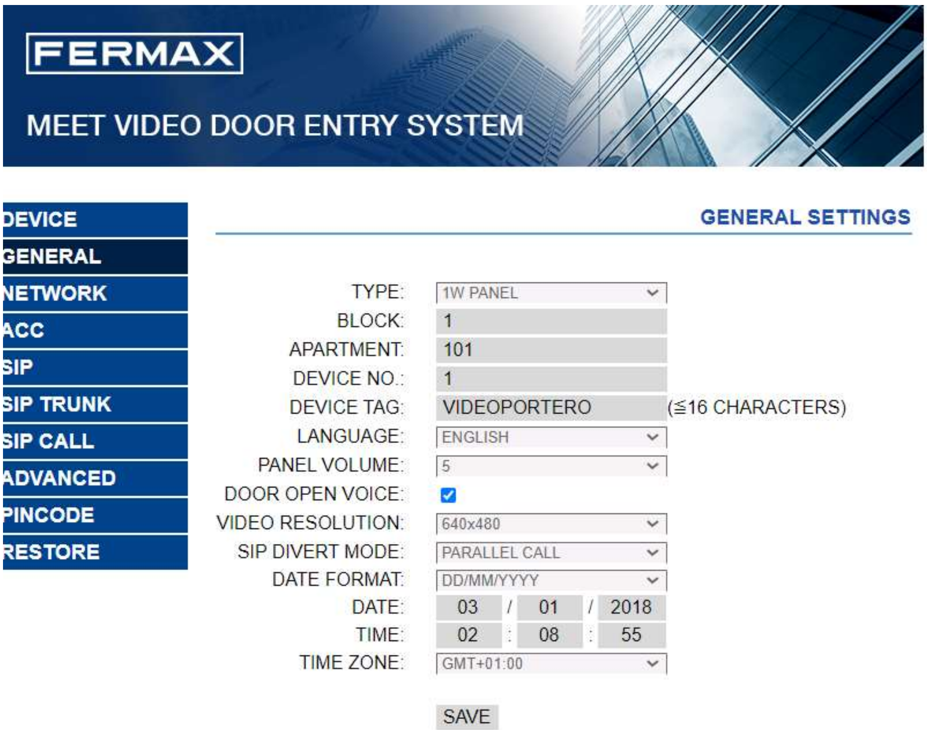
In the NETWORK tab, change the IP to the desired one, in this case 192.168.1.51 and click SAVE. Then we need to change the computer network ip address so that it is in the same network range 192.168.1.x

Now we can access again to the webserver introducing the new ip address 192.168.1.51 in the web browser and we will see again the MILO setup page.

In the GENERAL tab, we will configure that is a 1 single pushbutton panel that calls an apartment (that really do not exist) in the block 1, apartment 101. The pushbutton now is mapped to apartment 101.

It should be selected the TYPE 1W PANEL .

The DEVICE NO will be 1 , to indicate that this is the first panel on the installation. If there were more, we will add consecutive numbers.



Now it is required to associate the pushbutton to an IP address or sip extension to call. We will go to SIP CALL tab and map the intercom pushbutton (assigned previously to apartment 101) to an IP address or SIP extension, that will be the **YEALINK** telephone.

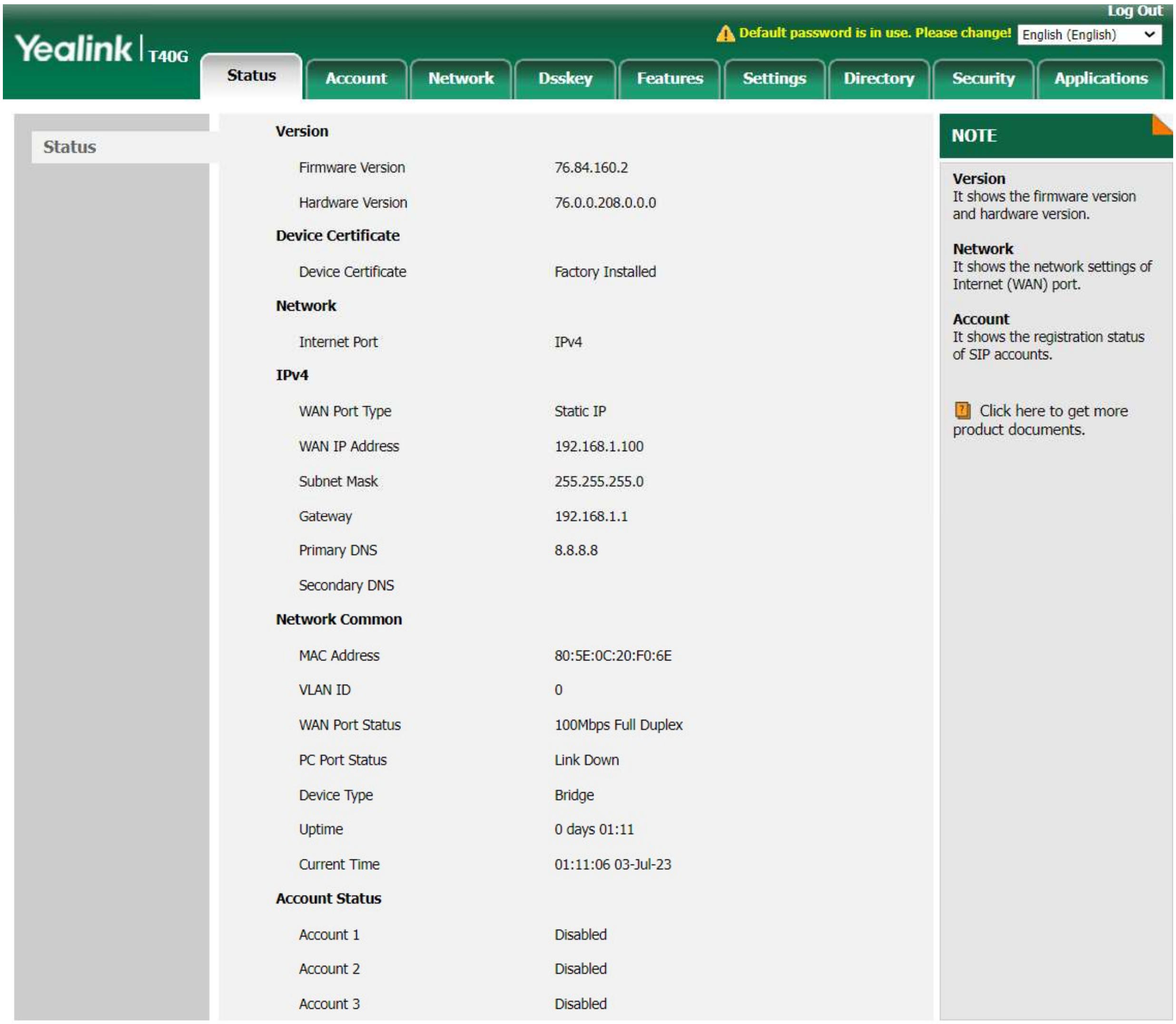
In the example, the field APARTMENT will be then 101, and the NUMBER will be the ip call destination, that in this case will be a sip call to ip address 192.168.1.100, assigned to YEALINK telephone. The syntaxis to be used is the following: **sip:192.168.14.1**

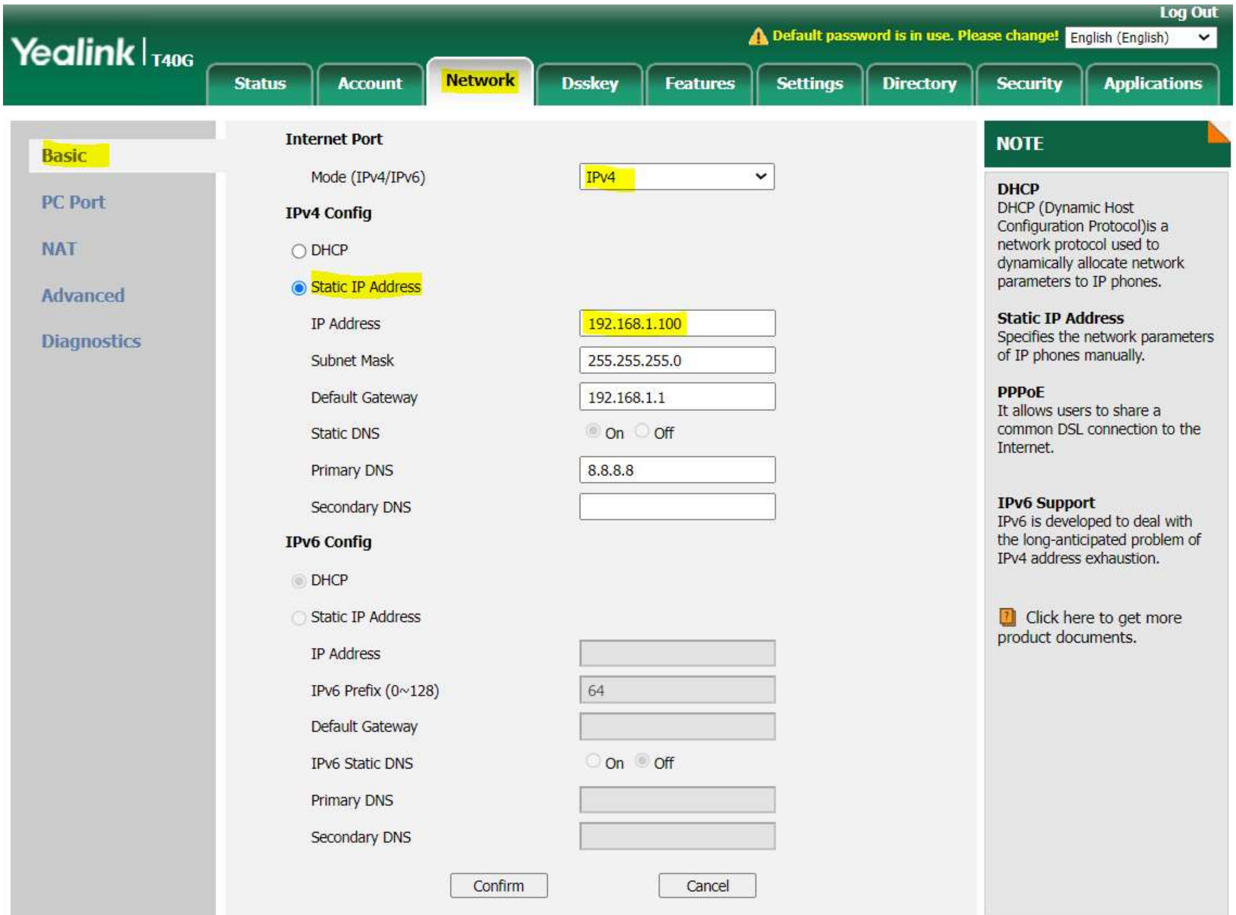
REMARK: These telephones are SIP, so they can be also registered as an extension of a SIP server. If that was the case, and assuming that the server ip would be 192.168.1.199 and the phone extension registered 1122, the call destination to be used would be then sip:1122@192.168.1.199 . Where this syntaxis corresponds to sip:sip_extension@server_ip_address

YEALINK SIP TELEPHONE CONFIGURATION

The YEALINK SIP telephone comes with its default ip.

We need to set the ip address in the desired address (in this case 192.168.1.100). This telephone also has a webserver embedded, so we will connect to it using the same web browser Google Chrome. To access we will use the credentials **user: admin** **pass: admin**


Yealink T40G
Log Out
Default password is in use. Please change!
English (English)
Status
Account
Network
Dsskey
Features
Settings
Directory
Security
Applications
Status
Version
Firmware Version 76.84.160.2
Hardware Version 76.0.0.208.0.0.0
Device Certificate
Device Certificate Factory Installed
Network
Internet Port IPv4
IPv4
WAN Port Type Static IP
WAN IP Address 192.168.1.100
Subnet Mask 255.255.255.0
Gateway 192.168.1.1
Primary DNS 8.8.8.8
Secondary DNS
Network Common
MAC Address 80:5E:0C:20:F0:6E
VLAN ID 0
WAN Port Status 100Mbps Full Duplex
PC Port Status Link Down
Device Type Bridge
Uptime 0 days 01:11
Current Time 01:11:06 03-Jul-23
Account Status
Account 1 Disabled
Account 2 Disabled
Account 3 Disabled
NOTE
Version
It shows the firmware version and hardware version.
Network
It shows the network settings of Internet (WAN) port.
Account
It shows the registration status of SIP accounts.
Click here to get more product documents.



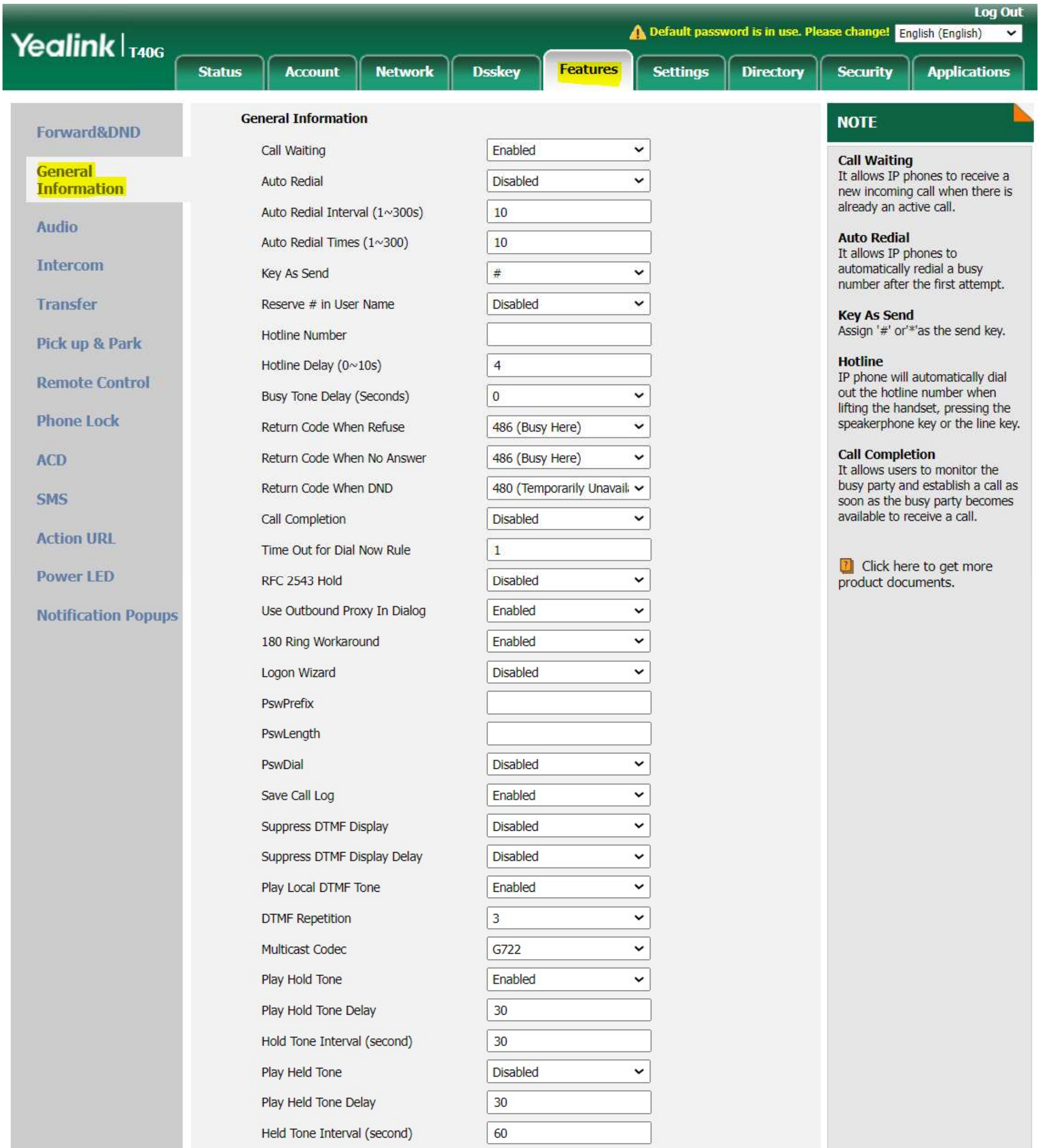
As default, YEALINK telephones do not receive direct calls from other SIP devices, so that we need to enable this feature in the menu

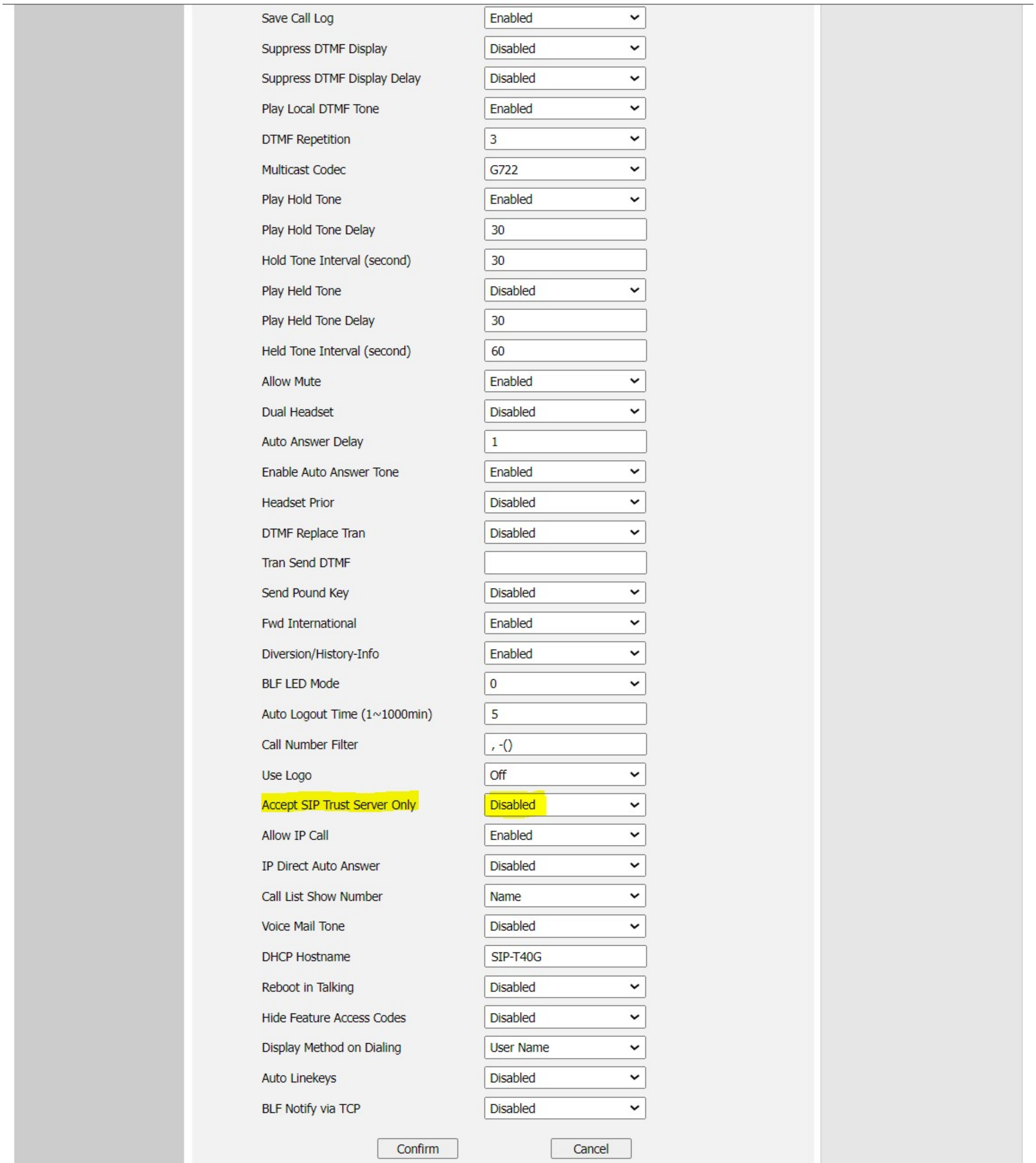
FEATURES → tab **GENERAL INFORMATION**

in the option

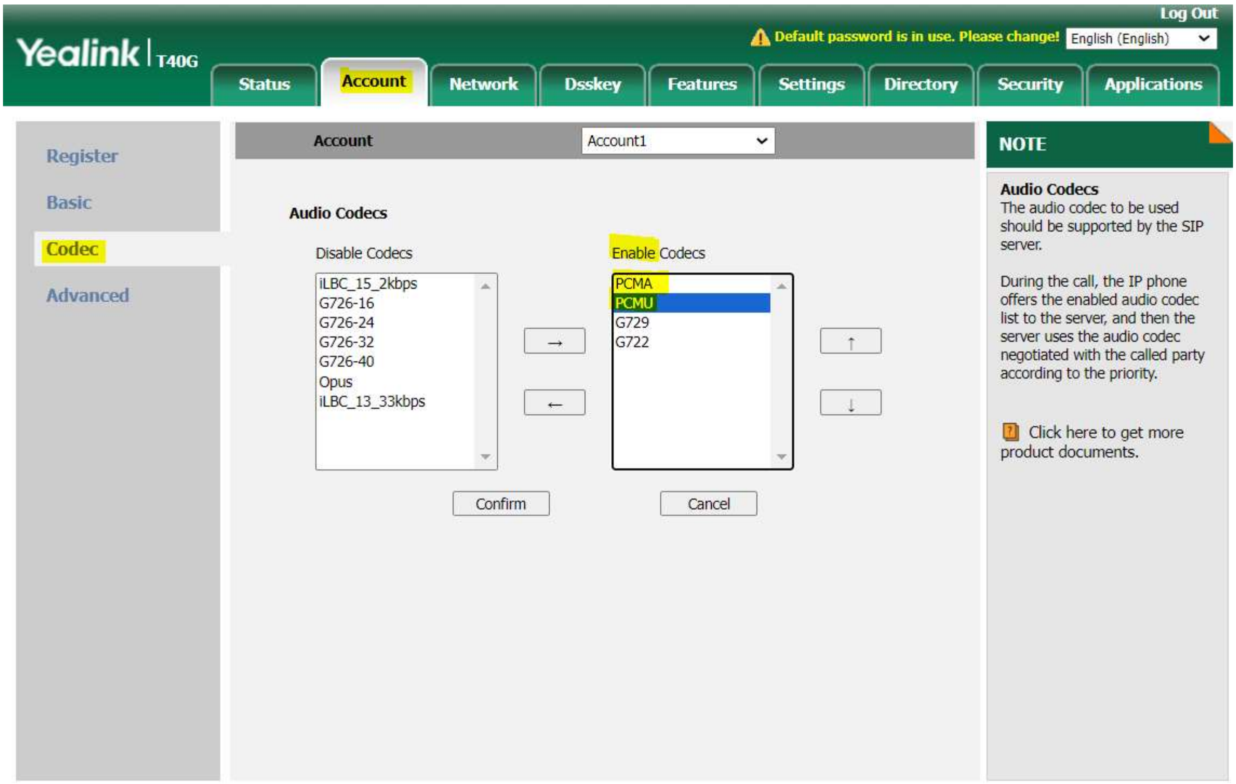
ACCEPT SIP TRUST SERVER ONLY → Disabled

The screens where to setup the YEALINK parameters are the following:


Yealink T40G
Log Out
Default password is in use. Please change!
English (English)
Status
Account
Network
Dsskey
Features
Settings
Directory
Security
Applications
Forward&DND
General Information
Audio
Intercom
Transfer
Pick up & Park
Remote Control
Phone Lock
ACD
SMS
Action URL
Power LED
Notification Popups
General Information
Call Waiting Enabled
Auto Redial Disabled
Auto Redial Interval (1~300s) 10
Auto Redial Times (1~300) 10
Key As Send #
Reserve # in User Name Disabled
Hotline Number
Hotline Delay (0~10s) 4
Busy Tone Delay (Seconds) 0
Return Code When Refuse 486 (Busy Here)
Return Code When No Answer 486 (Busy Here)
Return Code When DND 480 (Temporarily Unavail
Call Completion Disabled
Time Out for Dial Now Rule 1
RFC 2543 Hold Disabled
Use Outbound Proxy In Dialog Enabled
180 Ring Workaround Enabled
Logon Wizard Disabled
PswPrefix
PswLength
PswDial Disabled
Save Call Log Enabled
Suppress DTMF Display Disabled
Suppress DTMF Display Delay Disabled
Play Local DTMF Tone Enabled
DTMF Repetition 3
Multicast Codec G722
Play Hold Tone Enabled
Play Hold Tone Delay 30
Hold Tone Interval (second) 30
Play Held Tone Disabled
Play Held Tone Delay 30
Held Tone Interval (second) 60
NOTE
Call Waiting
It allows IP phones to receive a new incoming call when there is already an active call.
Auto Redial
It allows IP phones to automatically redial a busy number after the first attempt.
Key As Send
Assign '#' or'*'as the send key.
Hotline
IP phone will automatically dial out the hotline number when lifting the handset, pressing the speakerphone key or the line key.
Call Completion
It allows users to monitor the busy party and establish a call as soon as the busy party becomes available to receive a call.
Click here to get more product documents.



It is convenient to confirm that the MEET compatible codecs are selected. We can do it on the SIP screen. Main codecs are PCMU and PCMA.



With the previous configuration we can make a call from the intercom panel to the telephone.

The door lock release can be done using either "*" or "#" from the telephone keypad, during the call.

VIDEO

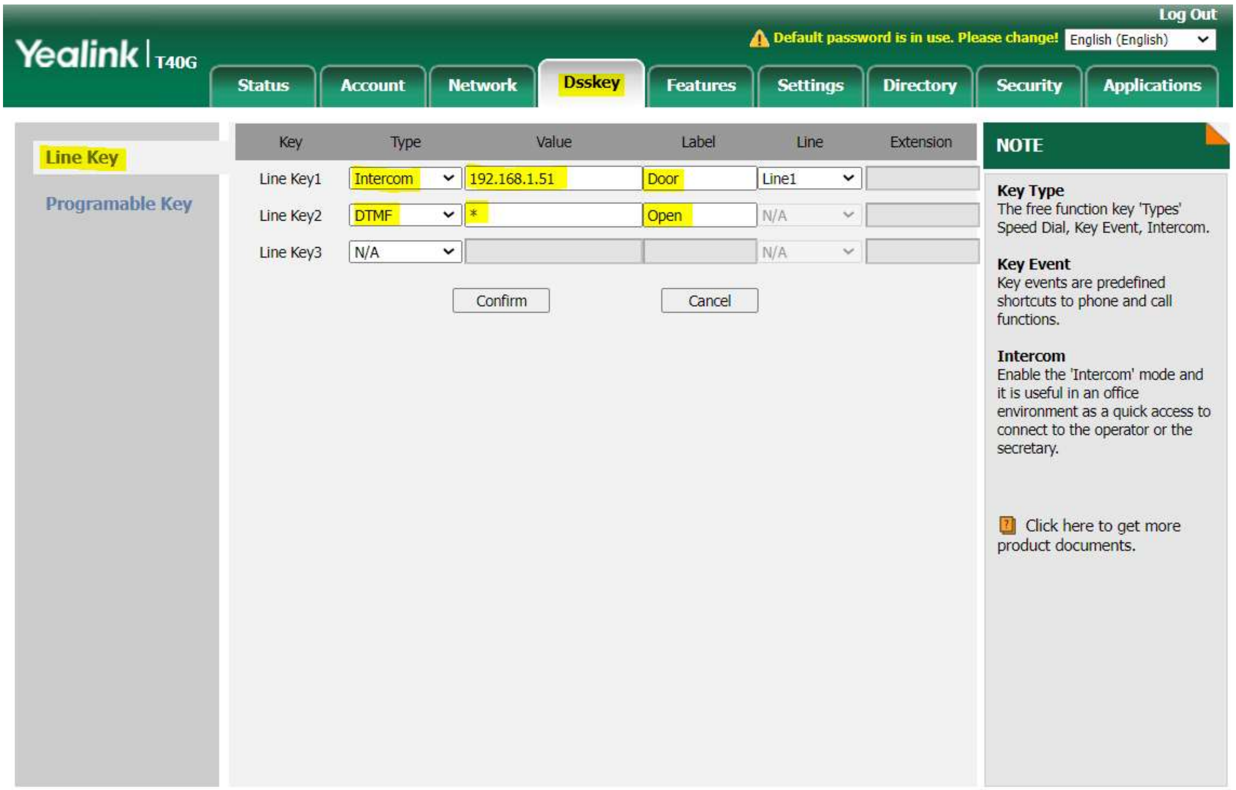
Currently YEALINK telephones on this case study do not allow video H.264 reception, which is the video sent by MEET FERMAX intercom panels. It is required to get an DOORPHONE INTEGRATION compatible YEALINK telephone that accepts H.264 video stream reception. Some of the telephones that are compatible with this feature are T58W or T57W.

ADVANCED FUNCTIONS

It is possible to call from the telephone to the panel. For that reason, it is quite useful to setup a direct access on the telephone. In particular, YEALINK TG40 allows to have direct accesses on the auxiliar display.

Additionally, for more usability, it is possible to set up a function key labeled to open the door, in this way even staff that is not familiar with the telephone will be able to identify the way to open the door.

For that it is recommended that a direct access is added to the directory menu.



To label properly and assign the functionality to the quick access function keys, we will access the "Dsskey " and set it up the desired keys.

To **call the outdoor intercom** panel we will choose

Type: Intercom Value: 192.168.1.51 Label:Door Line: Line1

Where "Value" is the IP of the intercom and "Label" is the text we want to be shown in the function key display.

To **open the door lock during the call** we will choos

Type: DTMF Value:* Label: Open Line

Where "Value" is the DTMF tone to open the door, that in MEET can be "*" or "#" to active the relay in the intercom panel. In case there is a secondary relay module connected to the panel, then this additional relay can be opened using "0" number. "Label" is the text we want to be shown in the function display.